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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/866,596	05/30/2001	Jebu Jacob Rajan	1263.1778	4950

5514 7590 01/13/2005

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EXAMINER	
VO, HUYEN X	
ART UNIT	**PAPER NUMBER**
2655	

DATE MAILED: 01/13/2005

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	09/866,596	RAJAN, JEBU JACOB
	Examiner	**Art Unit**
	Huyen Vo	2655

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *01 September 2004*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-53 and 56-76* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-53 and 56-76* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *19 September 2001* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments, see amendment, filed 9/7/2004, with respect to the rejection(s)of claim(s) 1-55 under U.S.C 103(a) have been fully considered and are persuasive. Therefore, the rejection has been withdrawn. However, upon further consideration, a new ground(s) of rejection is made in view of Haimi-Cohen (US 6374221) and Handel et al. (US 6324502).

Claim Rejections - 35 USC § 102

2. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless – (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

3. Claims 49-53 and 75-76 are rejected under 35 U.S.C. 102(e) as being anticipated Haimi-Cohen (US 6374221).

4. Regarding claims 50-53 and 76, Haimi-Cohen disclose an audio comparison apparatus comprising: a memory for storing a predetermined function which gives, for a given set of audio signal values, a probability density for parameters of a predetermined audio model which is assumed to have generated the set of audio signal values, the probability density defining, for a given set of model parameter values, the probability that the predetermined audio model has those parameter values, given that the model is

assumed to have generated the set of audio signal values (*col. 7, line 32 to col. 8, line 20*); means for receiving a set of audio signal values representative of an input audio signal (*input of element 51 in figure 5*); means for applying the set of received audio signal values to said stored function to give the probability density for said model parameters for the set of received audio signal values (*col. 7, line 32 to col. 8, line 20, received signal is applied to a stored function to derive HMM, wherein each state of the HMM is represented by probabilistic distribution*); means for processing said function, with said set of received audio signal values applied, to derive samples of parameter values from said probability density (*col. 7, line 32 to col. 8, line 20, received signal is applied to a stored function to derive HMM, wherein each state of the HMM is represented by probabilistic distribution*); means for analyzing at least some of said derived samples of parameter values to determine parameter values that are representative of the set of received audio signal values (*col. 7, line 32 to col. 8, line 20, received signal is applied to a stored function to derive HMM, wherein each state of the HMM is represented by probabilistic distribution*); and means for comparing said determined parameter values with pre-stored parameter values to generate a comparison result (*element 53 of figure 5*).

5. Regarding claims 49 and 75, Haimi-Cohen disclose an apparatus for determining sets of parameter values representative of an input speech signal, the apparatus comprising: means for receiving a plurality of speech signal values representative of an input speech signal (*input of feature extractor 51 in figure 5*); means for dividing the

plurality of speech signal values into a succession of groups of speech signal values (*col. 7, lines 4-20, Frames*); and means for processing the speech signal values in each group to determine a set of parameter values representative of the speech signal values in the group (*feature extractors 51 in figure 5*); wherein said processing means comprises: a memory for storing data defining a predetermined function which gives, for a set of speech signal values of a group, a probability density for parameters of a predetermined signal model which is assumed to have generated the speech signal values in the group, the probability density defining, for a given set of parameter values, the probability that the predetermined signal model has those parameter values, given that the model is assumed to have generated the speech signal values in the group (*col. 7, line 32 to col. 8, line 20, received signal is applied to a stored function to derive HMM, wherein each state of the HMM is represented by probabilistic distribution*); means for applying the set of speech signal values of a current group to said stored function to give the probability density for said model parameters for the current group (*col. 7, line 32 to col. 8, line 20, received signal is applied to a stored function to derive HMM, wherein each state of the HMM is represented by probabilistic distribution*); means for processing said function to derive samples of parameter values from said probability density for the current group (*col. 7, line 32 to col. 8, line 20, received signal is applied to a stored function to derive HMM, wherein each state of the HMM is represented by probabilistic distribution*); means for evaluating said probability density for the current group using one or more of said derived samples of parameter values for different numbers of parameter values to determine respective probabilities that the

predetermined signal model has those parameter values (*col. 8, lines 21-50, comparing with the stored models, represented by HMM, to determine if there is a match*); and means for processing at least some of said derived samples of parameter values and said evaluated probabilities to determine model parameters that are representative of the set of signal values in the current group (*col. 8, lines 1-50, comparing with the stored models, represented by HMM, to determine if there is a match*).

6. Claims 1, 11-19, 46, 48, 56, 61-65, 72, and 74 are rejected under 35 U.S.C. 102(e) as being anticipated by Handel et al. (US 6324502).

7. Regarding claims 1 and 56, Handel et al. disclose a speech processing apparatus comprising: means for receiving a set of speech signal values representative of a speech signal generated by a speech source as distorted by a transmission channel between the speech source and the receiving means (*elements 20-22 in figure 1*); a memory for storing data defining a predetermined function derived from a predetermined signal model which includes a first part having first parameters which models said source and a second part having second parameters which models said channel and said function being in terms of said first and second parameters and in terms of a set of speech signal values (*Elements 18 and 22 in figure 1*); means for applying said set of received signal values to said stored function (*the operation of figure 1, the received signal is passed through elements 18 and 22 for estimating signal parameters*); and means for processing said function with those values applied to obtain values of said first parameters that are representative of said speech generated by said

speech source before it was distorted by said transmission channel (*the operation of figure 1, the clean speech signal is output at the end representing the speech signal before distorted by the microphone and the transmission channel*).

8. Regarding claims 46 and 72, Handel et al. disclose a speech processing apparatus comprising: means for receiving a set of signal values representative of a speech signal generated by a speech source as distorted by a transmission channel between the speech source and the receiving means (*elements 20-22 in figure 1*); a memory for storing data defining a predetermined function derived from a predetermined signal model which includes a first part having first parameters which models said source and a second part having second parameters which models said channel, said function being in terms of said first and second parameters and being in terms of a set of raw speech signal values representative of speech generated by said source before being distorted by said transmission channel (*elements 34 in figures 1 and AR estimates for noise and speech 18 and 22*); means for processing said received set of signal values with initial estimates of said first and second parameters, to generate an estimate of the raw speech signal values corresponding to the received set of signal values (*elements 20, 30 and 32 in figure 1*); means for applying said set of received signal values and the estimated set of raw speech signal values to said function (*element 34 in figure 1 received raw speech signal values and the estimated speech signal values*); means for processing said function with those values applied to obtain values of said first parameters that are representative of said speech signal

generated by said speech source before it was distorted by said transmission channel (*output of Kalman filter 34 in figure 1*).

9. Regarding claims 48 and 74, Handel et al. disclose an apparatus for determining sets of parameter values representative of an input speech signal, the apparatus comprising: means for receiving a plurality of speech signal values representative of an input speech signal (*elements 20-22 in figure 1*); means for dividing the plurality of speech signal values into a succession of groups of speech signal values (*col. 2, lines 56-67, audio frames*); and means for processing the speech signal values in each group, to determine a set of parameter values representative of the speech signal values in each group (*Output of element 18 in figure 1*); wherein said determining means comprises means for varying the number of parameter values used to represent the speech signal values in each group (*Output of elements 20 and/or 30 in figure 1*).

10. Regarding claims 11 and 61, Handel et al. further disclose an apparatus, wherein said function is in terms of a set of raw speech signal values representative of speech generated by said source before being distorted by said transmission channel (*output of element 32 is used to control the Kalman Filter*), wherein the apparatus further comprises second processing means for processing the received set of signal values with initial estimates of said first and second parameters, to generate an estimate of the raw speech signal values corresponding to the received set of signal values (*elements 20, 26, and 30 in figure 1*) and wherein said applying means is operable to apply said

estimated set of raw speech signal values to said function in addition to said set of received signal values (*element 34 in figure 1*).

11. Regarding claims 12-13, Handel et al. further disclose an apparatus according to claim 11, wherein said second processing means comprises a simulation smoother (*element 32 in figure 1*), and a Kalman filter (*element 34 in figure 1*).

12. Regarding claims 14-17 and 62-64, Handel et al. further disclose an apparatus, wherein said receiving means is operable to receive a sequence of sets of signal values representative of a speech signal generated by a speech source as distorted by said transmission channel (*microphone 10 in figure 1*) and wherein said processing means is operable to obtain values of said first parameters for the speech within each set of signal values in said sequence (*elements 20 and 30-34 in figure 1*), and wherein said processing means is operable to use the values of said first parameters obtained during the processing of a preceding set of signal values as initial estimates for the values of said first parameters for a current set of signal values being processed (*col. 6, lines 30-39*), and wherein said sets of signal values in said sequence are non-overlapping (*col. 2, lines 56-67*), and wherein said processing means comprises means for varying the number of parameter values used to represent the speech signal within each set of signal values (*elements 20 and 26 in figure 1*).

13. Regarding claims 18-19 and 65, Handel et al. further disclose an apparatus, wherein said first part is an auto-regressive model and said first parameters comprise auto-regressive model coefficients (*element 18 in figure 1*), and wherein said second part is a moving average model and said second parameters comprise moving average model coefficients (*col. 6, lines 1-11*).

Claim Rejections - 35 USC § 103

14. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

15. Claims 2, 4-8, 23-24, 26-30, 33-45, 47, 57, 59, 66, 70-71, and 73 are rejected under 35 U.S.C. 103(a) as being unpatentable over Handel et al. (US 6324502) in view of Higgins et al. (US 6266633).

16. Regarding claims 23, 45, and 70-71, Handel et al. disclose a speech processing method comprising the steps of: receiving a set of signal values representative of a speech signal generated by a speech source as modified by a transmission channel between the speech source and the receiver (*elements 20-22 in figure 1*); storing data defining a predetermined function derived from a predetermined signal model which includes a first part having first parameters which models said source and a second part

having second parameters which models said channel, said function being in terms of said first and second parameters (*Elements 18 and 22 in figure 1*); applying said set of received signal values to said function (*the operation of figure 1, the received signal is passed through elements 18 and 22 for estimating signal parameters*); means for analyzing at least some of said derived samples to determine values of said first parameters that are representative of said speech signal generated by said source before it was modified by said transmission channel (*element 34 in figure 1*).

Handel et al. fail to specifically disclose the step of generating, for a given set of signal values, a probability density function which defines, for a given set of first parameters and second parameters, the probability that the predetermined signal model has those parameter values, given that the signal model is assumed to have generated the received set of signal values; and processing said function with those values applied to derive samples of at least said first parameters from said probability density function.

However, Higgins et al. teach the step of generating, for a given set of signal values, a probability density function which defines, for a given set of first parameters and second parameters, the probability that the predetermined signal model has those parameter values, given that the signal model is assumed to have generated the received set of signal values (*col. 1, lines 1-67*); and processing said function with those values applied to derive samples of at least said first parameters from said probability density function (*col. 1, lines 1-67*).

Since Handel et al. and Higgins et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art

at the time of invention to modify Handel et al. by incorporating the teaching of Higgins et al. in order to enhance speech signal for subsequent processing.

17. Regarding claims 47 and 73, Handel et al. disclose a speech processing apparatus comprising: means for receiving a set of signal values representative of a speech signal generated by a speech source as modified by a transmission channel between the signal source and the receiving means (*elements 20-22 in figure 1*); a memory for storing data defining a predetermined function derived from a predetermined signal model which includes a first part having first parameters which models said source and a second part having second parameters which models said channel, said function being in terms of said first and second parameters and being in terms of a set of raw speech signal values representative of a speech signal generated by said source before being modified by said transmission channel (*elements 18 and 22 in figure 1*); means for processing said received set of signal values with initial estimates of said first and second parameters, to generate an estimate of said set of raw speech signal values corresponding to the received set of signal values (*elements 18 and 22 in figure 1 and col. 8, lines 1-25*); means for applying said set of received signal values and the estimated set of raw speech signal values to said function (*Kalman filter 34 receive raw speech signal and estimated values*); and means for analyzing at least some of said derived samples to determine values of said first parameters that are representative of said speech signal generated by said speech source before it was modified by said transmission channel (*output of filter 34 is a clean speech signal*).

Handel et al. fail to specifically disclose the step of generating, for a given set of signal values, a probability density function which defines, for a given set of first parameters and second parameters, the probability that the predetermined signal model has those parameter values, given that the signal model is assumed to have generated the received set of signal values; and processing said function with those values applied to derive samples of at least said first parameters from said probability density function.

However, Higgins et al. teach the step of generating, for a given set of signal values, a probability density function which defines, for a given set of first parameters and second parameters, the probability that the predetermined signal model has those parameter values, given that the signal model is assumed to have generated the received set of signal values (*col. 1, lines 1-67*); and processing said function with those values applied to derive samples of at least said first parameters from said probability density function (*col. 1, lines 1-67*).

Since Handel et al. and Higgins et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Handel et al. by incorporating the teaching of Higgins et al. in order to enhance speech signal for subsequent processing.

18. Regarding claims 2, 24, and 57, Handel et al. fail to specifically disclose an apparatus, wherein said function generates, for a given set of received signal values, a probability density function which defines, for a given set of first and second parameters, the probability that the predetermined signal model has those parameter

values, given that the signal model is assumed to have generated the received set of signal values and wherein said processing means comprises means for drawing samples from said probability density function and means for determining said values of said first parameters that are representative of the speech from the drawn samples.

However, Higgins et al. teach that a function generates, for a given set of received signal values, a probability density function which defines, for a given set of first and second parameters, the probability that the predetermined signal model has those parameter values, given that the signal model is assumed to have generated the received set of signal values (*element 75 in figure 2B, pdf is needed before constructing histogram and/or referring to figure 4*) and wherein said processing means comprises means for drawing samples from said probability density function and means for determining said values of said first parameters that are representative of the speech from the drawn samples (*col. 7, lines 1-67, determining noise floor and channel response from the histogram before applying spectral subtraction to obtained a noise-suppressed speech signal*).

Since Handel et al. and Higgins et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Handel et al. by incorporating the teaching of Higgins et al. in order to enhance speech signal for subsequent processing.

19. Regarding claim 66, Handel et al. further disclose an apparatus, wherein said second part is a moving average model and said second parameters comprise moving average model coefficients (*col. 6, lines 1-11*).

20. Regarding claims 4-5, 26-27, and 59, An apparatus according to claim 2, wherein said processing means is operable to determine a histogram of said drawn samples and wherein said values of said first parameters are determined from said histogram (*col. 7, lines 1-67*), and wherein said processing means is operable to determine said values of said first parameters using a weighted sum of said drawn samples, and wherein the weighting is determined from said histogram (*col. 7, lines 1-67*).

21. Regarding claims 6-7 and 28-29, Handel et al. fail to disclose an apparatus, wherein said sampling means is operable to draw samples iteratively from said probability density function, and wherein said sampling means is operable to draw samples of both said first and second parameters. However, Higgins et al. further teach a sampling means is operable to draw samples iteratively from said probability density function (*the operation of the second diagram in figure* 3), and wherein said sampling means is operable to draw samples of both said first and second parameters (*the operation of the second diagram in figure* 3).

Since Handel et al. and Higgins et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Handel et al. by incorporating the teaching of

Higgins et al. in order to remove the noise from the received speech signal to enhance speech signal for subsequent processing.

22. Regarding claims 8 and 30, the modified Handel et al. fail to specifically disclose an apparatus, wherein said sampling means comprises a Gibbs sampler. However, a Gibbs sampler is well known to a person of ordinary skill in the art. The advantage of using a Gibbs sampler in the modified Handel et al. is facilitate the analysis of the received speech signal.

23. Regarding claim 33, Handel et al. further disclose an apparatus, wherein said function is in terms of a set of raw speech signal values representative of speech generated by said source before being distorted by said transmission channel (*output of element 32 is used to control the Kalman Filter*), wherein the apparatus further comprises second processing means for processing the received set of signal values with initial estimates of said first and second parameters, to generate an estimate of the raw speech signal values corresponding to the received set of signal values (*elements 20, 26, and 30 in figure 1*) and wherein said applying means is operable to apply said estimated set of raw speech signal values to said function in addition to said set of received signal values (*element 34 in figure 1*).

24. Regarding claims 34-35, Handel et al. further disclose an apparatus, wherein said second processing means comprises a simulation smoother (*element 32 in figure 1*), and a Kalman filter (*element 34 in figure 1*).

25. Regarding claims 36-39, Handel et al. further disclose an apparatus, wherein said receiving means is operable to receive a sequence of sets of signal values representative of a speech signal generated by a speech source as distorted by said transmission channel (*microphone 10 in figure 1*) and wherein said processing means is operable to obtain values of said first parameters for the speech within each set of signal values in said sequence (*elements 20 and 30-34 in figure 1*), and wherein said processing means is operable to use the values of said first parameters obtained during the processing of a preceding set of signal values as initial estimates for the values of said first parameters for a current set of signal values being processed (*col. 6, lines 30-39*), and wherein said sets of signal values in said sequence are non-overlapping (*col. 2, lines 56-67*), and wherein said processing means comprises means for varying the number of parameter values used to represent the speech signal within each set of signal values (*elements 20 and 26 in figure 1*).

26. Regarding claims 40-41, Handel et al. further disclose an apparatus, wherein said first part is an auto-regressive model and said first parameters comprise auto-regressive model coefficients (*element 18 in figure 1*), and wherein said second part is a

moving average model and said second parameters comprise moving average model coefficients (*col. 6, lines 1-11*).

27. Regarding claims 42-44, the modified Handel et al. fail to disclose an apparatus according to claim 1, further comprising means for comparing said determined parameter values with pre-stored parameter values to generate a comparison result, with pre-stored reference models to generate a recognition result, with pre-stored speaker models to generate a verification result. However, the examiner takes official notice that parameter matching, pattern matching, and/or model matching are well known speech/voice recognition system. The advantage of these matching is to determine the an appropriate match for the input command so that subsequent action can be take to perform the user's command.

28. Claims 3, 25, and 58 are rejected under 35 U.S.C. 103(a) as being unpatentable over Handel et al. (US 6324502) in view of Higgins et al. (US 6266633), and further in view of Haimi-Cohen (US 6374221).

29. Regarding claims 3, 25, and 58, the modified Handel et al. fail to disclose an apparatus according to claim 2, further comprising means for evaluating said probability density function for the set of received signal values using one or more of said drawn samples or parameter values for different numbers of parameter values, to determine respective probabilities that the predetermined signal model has those parameter

values and wherein said processing means is operable to process at least some of said drawn samples of parameter values and said evaluated probabilities to determine said values of said first parameters that are representative of the speech generated by said source before it was distorted by said transmission channel.

However, Haimi-Cohen teach means for evaluating said probability density function for the set of received signal values using one or more of said drawn samples or parameter values for different numbers of parameter values, to determine respective probabilities that the predetermined signal model has those parameter values (*col. 7, line 32 to col. 8, line 51, the received signal is applied to a stored function to derive HMM, wherein each state of the HMM is represented by probabilistic distribution*) and wherein said processing means is operable to process at least some of said drawn samples of parameter values and said evaluated probabilities to determine said values of said first parameters that are representative of the speech generated by said source before it was distorted by said transmission channel (*col. 7, line 32 to col. 8, line 51, the received signal is compared with pre-stored speech models*).

Since the modified Handel et al. and Haimi-Cohen are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Handel et al. by incorporating the teaching of Haimi-Cohen in order to recognize the received speech command so that appropriate service can be provided to the user.

30. Claims 9-10, 31-32, and 60 are rejected under 35 U.S.C. 103(a) as being unpatentable over Handel et al. (US 6324502) in view of Higgins et al. (US 6266633), and further in view of Boggs (US 4860360).

31. Regarding claims 9, 31, and 60, Handel et al. further disclose an apparatus according to claim 2, wherein said processing means comprises means for analyzing at least some of said drawn samples of parameter values to determine a measure of the variance of said samples (*output of Speech AR estimate 18 in figure 1*), but fail to specifically disclose an apparatus that further comprises means for outputting a signal indicative of the quality of the received set of signal values in dependence upon said determined variance measure. However, Boggs teaches means for outputting a signal indicative of the quality of the received set of signal values in dependence upon said determined variance measure (*col. 6, lines 31-37*).

Since the modified Handel et al. and Boggs are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Handel et al. by incorporating the teaching of Boggs in order to indicate to the user that the quality of service was degraded so that appropriate action can be taken to correct the problem.

32. Regarding claims 10 and 32, Handel et al. fail to disclose an apparatus, wherein said probability density function is in terms of said variance measure and wherein said analyzing means is operable to draw samples of said variance measure from said

probability density function. However, Higgins et al. further teach that the probability density function is in terms of said variance measure and wherein said analyzing means is operable to draw samples of said variance measure from said probability density function (*col. 7, lines 1-67 and figure 4*).

Since Handel et al. and Higgins et al. are analogous art because they are from the same field of endeavor, it would have been obvious to one of ordinary skill in the art at the time of invention to further modify Handel et al. by incorporating the teaching of Higgins in order to perform noise suppression to enhance the signal for subsequent processing.

33. Claims 20-22 and 67-69 are rejected under 35 U.S.C. 103(a) as being unpatentable over Handel et al. (US 6324502).

34. Regarding claims 20-22 and 67-69, Handel et al. fail to disclose an apparatus according to claim 1, further comprising means for comparing said determined parameter values with pre-stored parameter values to generate a comparison result, with pre-stored reference models to generate a recognition result, with pre-stored speaker models to generate a verification result. However, the examiner takes official notice that parameter matching, pattern matching, and/or model matching are well known speech/voice recognition system. The advantage of these matching is to determine the an appropriate match for the input command so that subsequent action can be take to perform the user's command.

Conclusion

The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Rajan et al. teach a Bayesian approach to parameter estimation and interpolation of time-varying autoregressive processes using the Gibbs sampler that is considered pertinent to the claimed invention.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Huyen Vo whose telephone number is 703-305-8665. The examiner can normally be reached on M-F, 9-5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Doris To can be reached on 703-305-4827. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Examiner Huyen X. Vo

January 7, 2005

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SUSAN MCFADDEN
PRIMARY EXAMINER